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Serial No.: 10/789,425
Notice of Appeal Dated: February 22, 2012

PATENT
PRN-012

**IN THE UNITED STATE PATENT AND TRADEMARK OFFICE
BEFORE THE BOARD OF PATENT APPEALS AND INTERFERENCES**

In re application of: Michael L. Petroff
For: SPEAKER SYSTEMS AND METHODS HAVING AMPLITUDE
AND FREQUENCY RESPONSE COMPENSATION
Serial No. 10/789,425
Filed February 27, 2004
Art Unit 2614
Examiner George C. Monikang
Attorney Docket No. PRN-012
Confirmation No. 5685

APPEAL BRIEF

ON APPEAL FROM GROUP ART UNIT 2614

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Commissioner for Patents
P.O. Box 1450
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Sir:

This Appcal Brief is submitted both in support of the Notice of Appeal filed February 22, 2012 and in response to the Office Action dated September 16, 2011.

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I. REAL PARTY IN INTEREST

The real party in interest is Thomson Licensing/Technicolor, the owner of 100% interest of assignee of record, PREMIER RETAIL NETWORKS, INC..

II. RELATED APPEALS AND INTERFERENCES

Appellant is not aware of any pending appeals, judicial proceedings, or interferences which may be related to, directly affect, be directly affected by, or have a bearing on the Board's decision in the pending appeal.

III. STATUS OF CLAIMS

The status of the claims in the present application is provided immediately below as follows:

- a) Claims 1-28 are pending in this application.
- b) Claims 1, 7, 12, 18, 22 and 25 are the independent claims.
- c) Claims 1-28 stand rejected in an Office Action dated September 16, 2011.
- d) Claims 1-28 are the subject of this appeal

IV. STATUS OF AMENDMENTS

All amendments filed to date in this application have been entered and there are no other claim amendments pending. The claims listed in the Claims Appendix (i.e., Section VIII) of this Appeal Brief correspond to the claims submitted in Appellant's Response to an Office Action which was filed on August 17, 2010. Those claim amendments were reviewed and entered by the Examiner.

V. SUMMARY OF CLAIMED SUBJECT MATTER

It should be explicitly noted that it is not Appellant's intention that the currently claimed or described embodiments be limited solely to operation within the illustrative embodiments identified below. Furthermore, citations to exemplary descriptions of illustrative embodiments are provided below in association with portions of the claims, which are related to the identified illustrative embodiments, entirely for compliance with, and in satisfaction of, the requirements for filing this

appeal. There is no intention to read any further interpreted limitations into the claims as presented. Moreover, it will be appreciated that additional exemplary descriptions, though not cited herein, may be present in this patent application.

The claimed invention, as recited in claim 1 is directed to a speaker system for providing enhanced intelligibility of a reproduced audio program signal in the presence of ambient noise (*primarily Figure and Abstract*) comprising: means for receiving the reproduced audio program signal (*Figure 2, element P_2 and page 5, paragraph [00018]*); a microphone for monitoring at least ambient noise signals and for providing a microphone output signal (*Figure 2, element MIC1 and paragraphs [00018] and [00019]*); means for enabling the microphone output signal during first increments of time when the reproduced audio program signal is substantially off, and disabling the microphone output signal during second increments of time when the reproduced audio program signal is on, such that the microphone output signal includes ambient noise signal components without including reproduced audio program signal components (*Figure 2, elements D2 and E1 and paragraph [00019]*); and a signal process, in communication with the means for receiving and the means for enabling/disabling, for applying a first transfer function to the reproduced audio program signal, the first transfer function incrementally increasing gain adjustments to the reproduced audio program signal as a function of an increasing average amplitude of the microphone output signal over a period of time during which said microphone output signal is enabled, and incrementally decreasing gain adjustments to the reproduced audio program signal as a function of a decreasing average amplitude of the microphone output signal over a period of time during which said microphone output signal is enabled (*Figure 2, element P_2 and f_2 , and paragraphs [00018] and [00019]*).

The claimed invention, as recited in claim 7 is directed to a speaker system for providing enhanced intelligibility of a reproduced audio program signal in the presence of ambient noise which includes a means for receiving the reproduced audio program signal (*Figure 3 and element P_3 and paragraph [00020]*); a microphone for monitoring ambient noise signals and for providing a microphone output signal (*Figure 3, element MIC1 and paragraph [00020]*); means for enabling the microphone output signal during first increments of time when the reproduced audio program signal is substantially off, and disabling the microphone output signal during second increments of time when the reproduced audio program signal is on, such that the microphone output signal

includes ambient noise signal components without including reproduced program signal components (*Figure 3, elements D2 and E1 and paragraph [00020]*); and a signal processor, in communication with the means for receiving and the means for enabling/disabling for applying a transfer function to the reproduced audio program signal, the transfer function incrementally increasing high frequency response of the reproduced audio program signal as a function of a decreasing amplitude of the microphone output signal averaged over a period of time during which said microphone output signal is enabled, and vice versa, wherein the signal process output signal is maintained during such times as the microphone output signal is disabled (*Figure 3, element P₃ and f₃, and paragraphs [00020] and [00021]*).

The claimed invention, as recited in claim 12 is directed to a speaker system for providing enhanced intelligibility of a reproduced audio program signal in the presence of ambient noise which includes a means for receiving the reproduced audio program signal (*Figure 4 and element P₄ and paragraph [00022]*); a microphone for monitoring ambient noise signals and for providing a microphone output signal (*Figure 4, element MIC1 and paragraph [00022]*); means for enabling the microphone output signal during first increments of time when the reproduced audio program signal is substantially off, and disabling the microphone output signal during second increments of time when the reproduced audio program signal is on, such that the microphone output signal includes ambient noise signal components without including reproduced program signal components (*Figure 4, elements D2 and E1 and paragraph [00022]*); and a signal processor, in communication with the means for receiving and the means for enabling/disabling, including a first transfer function and a second transfer function having at least one signal processor output signal, wherein the first transfer function provides incrementally increasing gain adjustments to the reproduced audio program signal as a function of an increasing average amplitude of the microphone output signal over a period of time during which said microphone output signal is enabled, and vice versa; the second transfer function provides incrementally increasing high frequency response of the reproduced audio program signal as a function of a decreasing average amplitude of the microphone output signal over a period of time during which said microphone output signal is enabled, and vice versa; and the at least one signal processor output signal is maintained during such time as the microphone output signal is disabled (*Figure 4, element P₄ and f₂ and f₃, and paragraphs [00022] and [00023]*).

The claimed invention, as recited in claim 18 is directed to a method for providing enhanced intelligibility of a reproduced audio program signal in the presence of ambient noise corresponding

to system claim 1; the claimed invention, as recited in claim 22 is directed to a method for providing enhanced intelligibility of a reproduced audio program signal in the presence of ambient noise corresponding to system claim 7; and the claimed invention, as recited in claim 25 is directed to a method for providing enhanced intelligibility of a reproduced audio program signal in the presence of ambient noise corresponding to system claim 12.

VI. GROUND OF REJECTION TO BE REVIEWED ON APPEAL

Certain prior art rejections for this application are based on the following references: AAPA, fig. 1; paragraphs [0002-00006], U.S. Patent Publication No. 20030123680 to Lee et al. (hereinafter referenced as "Lee"); U.S. Patent 4,554,533 to Bosnak (hereinafter referenced as "Bosnak"); U.S. Patent 5,588,065 to Tanaka et al. (hereinafter referenced as "Tanaka").

The grounds of rejection for this application for which review is sought in this appeal are presented below as follows:

1. Whether claims 1-2, 7, 12-13, 18-19, 22 and 25-26 are properly rejected by the USPTO under 35 U.S.C. §103(a) as being unpatentable over AAPA in view of Lee.
2. Whether claims 3, 5, 8, 10, 14, 16, 20-21, 23-24, and 27-28 are properly rejected by the USPTO under 35 U.S.C. §103(a) as being unpatentable over AAPA and Lee as applied to claim 1, in view of Bosnak.
3. Whether claims 4, 6, 9, 11, 15 and 17 are properly rejected by the USPTO under 35 U.S.C. §103(a) as being unpatentable over AAPA, Lee and Bosnak as applied to claim 3, in view of Tanaka.
4. Regarding the grouping of the claims, claims 2-6 stand or fall with claim 1, claims 8-11 stand or fall with claim 7, claims 13-17 stand or fall with claim 12, claims 19-21 stand or fall with claim 18, claims 23-24 stand or fall with claim 22 and claims 26-28 stand or fall with claim 25.

VII. ARGUMENT

Appellant respectfully traverses the rejections in accordance with the detailed arguments set forth below.

**1. CLAIMS 1-2, 7, 12-13, 18-19, 22 AND 25-26 ARE IMPROPERLY
REJECTED BY THE USPTO UNDER 35 U.S.C. §103(a) AS BEING
UNPATENTABLE OVER AAPA IN VIEW OF LEE.**

Claims 1-2, 7, 12-13, 18-19, 22 and 25-26 are patentable over AAPA in view of Lee, as the references fail to disclose or render obvious at least “means for enabling the microphone output signal during first increments of time when the reproduced audio program signal is substantially off, and disabling the microphone output signal during second increments of time when the reproduced audio program signal is on, such that the microphone output signal includes ambient noise signal components without including reproduced audio program signal components” as taught and claimed by the Appellant.

Claimed subject matter is unpatentable under 35 U.S.C. 103(a) “if the differences between the subject matter sought to be protected and the prior art are such that the subject matter as a whole would have been obvious at the time the invention was made to a person having ordinary skill in the art to which said subject matter pertains.” *KSR International Co. v. Teleflex, Inc.*, 127 S.Ct.1727, 1734 (2007) (quoting 35 U.S.C. 103(a)). The Appellant submits that the differences between the claimed invention of the Appellant and the cited References are not obvious and were not obvious at the time the invention was made to a person having ordinary skill in the art. More specifically, the Appellant’s Claim 1 recites:

A speaker system providing enhanced intelligibility of a reproduced audio program signal in the presence of ambient noise, the speaker system comprising:

means for receiving the reproduced audio program signal;

a microphone for monitoring at least ambient noise signals and for providing a microphone output signal;

means for enabling the microphone output signal during first increments of time when the reproduced audio program signal is substantially off, and disabling the microphone output signal during second increments of time when the reproduced audio program signal is on, such that the microphone output signal includes ambient noise signal components without including reproduced audio

program signal components; and

a signal process, in communication with the means for receiving and the means for enabling/disabling, for applying a first transfer function to the reproduced audio program signal, the first transfer function incrementally increasing gain adjustments to the reproduced audio program signal as a function of an increasing average amplitude of the microphone output signal over a period of time during which said microphone output signal is enabled, and incrementally decreasing gain adjustments to the reproduced audio program signal as a function of a decreasing average amplitude of the microphone output signal over a period of time during which said microphone output signal is enabled.

It is respectfully asserted that none of AAPA, Lee, Bosnak, or Tanaka, alone or in combination, disclose, teach or suggest at least:

“means for enabling the microphone output signal during first increments of time when the reproduced audio program signal is substantially off, and disabling the microphone output signal during second increments of time when the reproduced audio program signal is on, such that the microphone output signal includes ambient noise signal components without including reproduced audio program signal components”

as claimed in the Appellant’s claim 1.

As admitted in a previous Office Action dated April 28, 2010, “AAPA fails to disclose incrementally increasing/ decreasing the gain adjustments.” (Office Action, page 4). Furthermore, as argued in response to several previous Office Actions, the Appellant submits that AAPA does not teach, show, or suggest the claimed limitation of “means for enabling the microphone output signal during first increments of time when the reproduced audio program signal is substantially off, and disabling the microphone output signal during second increments of time when the reproduced audio program signal is on” as taught and claimed by the Appellant.

More specifically, as stated above the Appellant submits that AAPA does not teach, show, or suggest the claimed limitations of “means for **enabling the microphone output signal** during first increments of time **when the reproduced audio program signal is substantially off**” and “the microphone output signal includes ambient noise signal components without including reproduced audio program signal components”. (Emphasis added). That is, the Appellant in the Specification teaches that the intelligibility of a reproduced program signal in the presence of widely varying

ambient noise levels is substantially enhanced by a signal process with processing functions that are incremental, as opposed to contiguous, such that the volume of the reproduced sound does not change too frequently as a consequence of rapidly occurring large changes in the ambient noise. Such signal process can provide increasing gain and increasing high frequency response of the program signal as a function of decreasing amplitude of a microphone output signal comprising of at least ambient noise signal components without reproduced program signal components by enabling such microphone output signal only while the program signal is substantially off, which typically occurs between audio or audio/video advertisements. (See Appellant's Specification, paragraph [0017]). The Appellant in the Specification further teaches that according to at least one embodiment of the Appellant's invention, these substantially off periods can be, for example as short as about 100 milliseconds (ms), but typically might be between about 100 to about 1000 ms. (See Appellant's Specification, paragraph [0019]).

In contrast to the invention of the Appellant, AAPA clearly states that "S2 transfers to signal output o of electronic switch E1, providing signal process control signal S3 only when **DC program signal S4 is in the off state, which occurs when S_{in} is below a minimum threshold level.**" (Emphasis added). *See AAPA paragraph [0005]*. In other words, in the AAPA the reproduced audio program signal (i.e., S_{in} in Figure 1) is not off. It is simply less than some threshold. The threshold is not defined as being at or near a zero signal value. That is, in AAPA, the signal that is off is S4 which is an output from DC detector D2. Since the microphone output via S3 in AAPA is switched to the control of the transfer function when the signal S4 is below a minimum threshold level, it is clear that the ambient noise signal will include a reproduced audio program signal. This occurs because the reproduced audio program signal in AAPA of Figure 1 is not off when the microphone output is applied. Since the reproduced audio signal in AAPA is only below a threshold level, then it is reasonable to believe that the reproduced audio signal exists at some amplitude level that allows it to be present in the microphone output signal. Obviously, this condition is contrary to the claimed limitation which state that ambient noise signal components [are present] without including reproduced audio program signal components.

Thus, the Appellant submits that AAPA does not teach the limitations of the independent claims and specifically at least "means for **enabling the microphone output signal** during first increments of time **when the reproduced audio program signal is substantially off**" and "the

microphone output signal includes ambient noise signal components without including reproduced audio program signal components” as taught in the Appellant’s Specification and claimed by at least the Appellant’s independent claims and specifically claim 1.

In addition, the Appellant submits that AAPA also fails to teach or suggest “a signal process, in communication with the means for receiving and the means for enabling/disabling, for applying a first transfer function to the reproduced audio program signal, the first transfer function incrementally increasing gain adjustments to the reproduced audio program signal as a function of an increasing average amplitude of the microphone output signal over a period of time during which said microphone output signal is enabled, and incrementally decreasing gain adjustments to the reproduced audio program signal as a function of a decreasing average amplitude of the microphone output signal over a period of time during which said microphone output signal is enabled” as taught and claimed by the Appellant.

More specifically, the invention of the Appellant includes a signal process for applying a first transfer function incrementally increasing gain adjustments to a reproduced audio program signal as a function of an increasing average amplitude of the microphone output signal over a period of time during which said microphone output signal is enabled, and incrementally decreasing gain adjustments to the reproduced audio program signal as a function of a decreasing average amplitude of the microphone output signal over a period of time during which said microphone output signal is enabled.

For example, with respect to FIG. 2, the Appellant teaches a first embodiment of an audio/video speaker system in which a program input signal $S_{sub.in}$ is applied to signal input s of signal process $P_{sub.2}$. In the embodiment of FIG. 2, $P_{sub.2}$ output port o provides signal process output signal $S5$. $P_{sub.2}$ introduces transfer function $f_{sub.2}$ providing incrementally increasing gain, for example, in steps of about 1 dB to about 10 dB, to $S_{sub.in}$ as a function of increasing amplitude of a signal process control signal, and vice versa, described below. The Appellant teaches that this transfer function $f_{sub.2}$ can, for example, be a non-linear equation of the form $f_{sub.2}(S_{sub.an}) = (S_{sub.in} \cdot S_{sub.an})^I$, where $S_{sub.an}$ is the ambient noise signal amplitude in increments of, for example, about 1 dB to about 10 dB. The Appellant teaches that microphone MIC1 provides an output signal $S1$ applied to level detector D1, which provides an output DC

microphone signal S2 applied to signal input s of electronic switch E1. S.sub.in is also applied to level detector D2, which provides an output DC program signal S4 applied to control input c of electronic switch E1. S2 transfers to signal output o of electronic switch E1, providing signal process control signal S3 only when DC program signal S4 is in the off state, which occurs when S.sub.in is off or substantially off, for example, below a minimum threshold level. S3 is applied to control input c of signal process P.sub.2 and thereby determines transfer function f.sub.2.

The Appellant teaches that the signal process P.sub.2 of FIG. 2 is maintained between such times as the microphone output signal is enabled (that is, switched through to the control input of the signal process) to provide continuing sound reproduction using the previously determined ambient noise level or average of levels. (See Appellant's Specification, paragraphs [0020]-[0021]).

As such, the Appellant submits that at least because in the claimed invention of the Appellant, the Appellant teaches and claims that the microphone output signal is enabled during a time when the reproduced audio program signal is off, the Appellant submits that the signal process of the Appellant's claimed invention occurs over a period of time during which said microphone output signal is enabled and at least because AAPA fails to teach or suggest enabling the microphone output signal during a time when the reproduced audio program signal is off, the Appellant further submits that AAPA also fails to teach or suggest "a signal process, in communication with the means for receiving and the means for enabling/disabling, for applying a first transfer function to the reproduced audio program signal, the first transfer function incrementally increasing gain adjustments to the reproduced audio program signal as a function of an increasing average amplitude of the microphone output signal over a period of time during which said microphone output signal is enabled, and incrementally decreasing gain adjustments to the reproduced audio program signal as a function of a decreasing average amplitude of the microphone output signal over a period of time during which said microphone output signal is enabled" as taught and claimed by the Appellant.

The Appellant further submits that the teachings of Lee absolutely fail to bridge the substantial gap between the teachings of AAPA and the invention of the Appellant. That is, the Appellant submits that there is absolutely no teaching or suggestion in Lee for incrementally increasing/ decreasing the gain adjustments, a "means for **enabling the microphone output signal** during first increments of time **when the reproduced audio program signal is substantially off**"

and that “the microphone output signal includes ambient noise signal components without including reproduced audio program signal components” and “a signal process, in communication with the means for receiving and the means for enabling/disabling, for applying a first transfer function to the reproduced audio program signal, the first transfer function incrementally increasing gain adjustments to the reproduced audio program signal as a function of an increasing average amplitude of the microphone output signal over a period of time during which said microphone output signal is enabled, and incrementally decreasing gain adjustments to the reproduced audio program signal as a function of a decreasing average amplitude of the microphone output signal over a period of time during which said microphone output signal is enabled” as taught and claimed by the Appellant.

In contrast to the invention of the Appellant, Lee teaches a volume control system and method of volume control for portable computer. That is Lee teaches a system for adjusting a sound signal output to a speaker of a portable computer based on an amount of background noise external to the personal computer. In Lee, a noise sensor senses the external noise and a micro controller calculates an average of the external noise for a predetermined period of time and adjusts a volume level controller based on a predetermined volume set up table which lists volume levels corresponding to levels of the external noise. Alternatively, the micro controller calculates an average noise level for a current predetermined period of time and adjusts the volume level controller based on comparing the current calculated average noise with a calculated average noise for a previous predetermined period of time. In the invention of Lee, if the external noise increases, the output sound signal is increased and if the external noise decreases, the output sound signal is decreased. (See Lee, Abstract).

The Examiner specifically cites Lee for teaching that a microphone monitors ambient noise, computes an average of the monitored ambient noise over a predetermined period of time, then incrementally increase/ decrease the volume of a sound signal accordingly. The Appellant disagrees. More specifically, the Examiner cites Table 1 and paragraphs 0028 and 0033 of Lee for attempting to teach that a microphone monitors ambient noise, computes an average of the monitored ambient noise over a predetermined period of time, then incrementally increase/ decrease the volume of a sound signal accordingly. Paragraphs 0028 and 0033 of Lee specifically recite:

“[0028] Further, the micro controller 12 comprises a volume set up table in which

volume levels corresponding to the noise levels graded by loudness of the external noise are listed. By way of example, the volume set up table is illustrated in Table 1.

[0033] Where the user selects the volume auto-control function through the user selection part 20 at operation S52, the micro controller 12 receives the external noise through the microphone 9, and calculates an average of the input external noise for a predetermined period of time at operation S54. The micro controller 12 determines the noise level on the basis of the calculated average of the external noise, and reads the volume level corresponding to the determined noise level from the volume set up table at operation S56. The micro controller 12 transmits the volume control signal based on the volume level from the volume set up table to the volume controller 14 at operation S5II, so that the sound is output through the speaker 7 on the basis of the read volume level from the volume set up table.”

As clear from paragraphs 0028 and 0033 of Lee, there is absolutely no teaching or suggestion in Lee that a microphone monitors ambient noise, computes an average of the monitored ambient noise over a predetermined period of time, then incrementally increase/ decrease the volume of a sound signal as alleged by the Examiner. More specifically, Table 1 merely illustrates a table that correlates ambient noise levels with a corresponding volume level for a volume controller. There is absolutely no teaching or suggestion in Lee that the volume levels of Table 1 are incremental and in fact seem to instead be continuous levels such that volume of a speaker is controlled in a continuous manner instead of incrementally as taught and claimed by the Appellant.

More specifically, paragraph [0006] of the Appellant’s Specification characterizes a problem of the prior art speaker systems as one wherein the conventionally compensated speaker output signal provides commensurately frequent and widely varying changes in sound levels that can be annoying to listeners. In contrast to continuous change systems, embodiments of the Appellant’s invention are directed to addressing a need to overcome this problem of the prior art speaker system by providing a speaker system providing direct, **but incremental**, amplitude compensation. From a simple analysis of these two sentences at the end of paragraph [0006], it is clear that the concept of “incremental” compensation of the frequent and widely varying changes in the ambient noise is definitely missing from the teachings of Lee because the Appellant states that an “incremental” approach is what is needed to overcome the problems in the prior art.

In describing embodiments of the invention, the Appellant in the Specification teaches that a solution to the prior art problem is to provide incremental compensation to the input audio signal. Incremental compensation is defined by the Appellant and distinguished expressly from continual or

continuous compensation as employed in the prior art speaker system of AAPA. In paragraph [0015] of the Appellant's Specification, the Appellant clearly draws a distinction between the terms "incremental" and "continuous" in an attempt to clarify the meaning of "incremental". In that paragraph, the Appellant states that the signal process with processing functions **"are incremental, as opposed to continuous."** [Emphasis supplied]. (See Appellant's response dated August 2009). Appellant unequivocally contrasts the terms "incremental" for the claimed invention with "continuous". By performing his functions in an "incremental" manner as opposed to a continuous manner of the prior art, the Appellant achieves the expressly desired solution to the prior art problems, namely, "that the volume of the reproduced sound does not change too frequently as a consequence of rapidly occurring large changes in the ambient noise". (See the Appellant's Specification at paragraph [0015] and compare with paragraph [0006]). The exemplary embodiments of various incremental adjustments are described and even claimed as being stepwise, such as in steps of about 1 dB to about 10 dB, in paragraphs [0018] and [0024] of the Appellant's Specification and in claims 2, 13, 19, and 26. The Appellant submits that in contrast to the invention of the Appellant, there is absolutely no teaching or suggestion in Lee that a microphone monitors ambient noise, computes an average of the monitored ambient noise over a predetermined period of time, then incrementally increase/ decrease the volume of a sound signal as alleged by the Examiner. The Appellant respectfully disagrees with the Examiner and submits that the teachings of Lee, which the Examiner used to re-open prosecution, suffer from the same deficiencies as every other Reference submitted by the Examiner and specifically Deville and Allen, which were the subject of a previous Appeal.

As noted above, the Appellant further submits that Lee also absolutely fails to teach or suggest a "means for **enabling the microphone output signal** during first increments of time **when the reproduced audio program signal is substantially off**" and that "the microphone output signal includes ambient noise signal components without including reproduced audio program signal components" and "a signal process, in communication with the means for receiving and the means for enabling/disabling, for applying a first transfer function to the reproduced audio program signal, the first transfer function incrementally increasing gain adjustments to the reproduced audio program signal as a function of an increasing average amplitude of the microphone output signal over a period of time during which said microphone output signal is enabled, and incrementally decreasing gain

adjustments to the reproduced audio program signal as a function of a decreasing average amplitude of the microphone output signal over a period of time during which said microphone output signal is enabled” as taught and claimed by the Appellant. In fact, the Examiner only cited Lee for teaching a system where a microphone monitors ambient noise, computes an average of the monitored ambient noise over a predetermined period of time, then incrementally increase/decrease the volume of a sound signal accordingly.

For at least the reasons recited above, the Appellant submits that AAPA and Lee, alone or in any allowable combination absolutely fail to teach or suggest at least incrementally increasing/decreasing the gain adjustments, a “means for **enabling the microphone output signal** during first increments of time **when the reproduced audio program signal is substantially off**” and that “the microphone output signal includes ambient noise signal components without including reproduced audio program signal components” and “a signal process, in communication with the means for receiving and the means for enabling/disabling, for applying a first transfer function to the reproduced audio program signal, the first transfer function incrementally increasing gain adjustments to the reproduced audio program signal as a function of an increasing average amplitude of the microphone output signal over a period of time during which said microphone output signal is enabled, and incrementally decreasing gain adjustments to the reproduced audio program signal as a function of a decreasing average amplitude of the microphone output signal over a period of time during which said microphone output signal is enabled” as taught and claimed by the Appellant. The failure of an asserted combination to teach or suggest each and every feature of a claim remains fatal to an obviousness rejection under 35 U.S.C. § 103. Section 2143.03 of the MPEP requires the “consideration” of every claim feature in an obviousness determination. To render a claim unpatentable, however, the Office must do more than merely “consider” each and every feature for this claim. Instead, the asserted combination of the patents must also teach or suggest *each and every claim feature*. See *In re Royka*, 490 F.2d 981, 180 USPQ 580 (CCPA 1974) (emphasis added) (to establish *prima facie* obviousness of a claimed invention, all the claim features must be taught or suggested by the prior art). Indeed, as the Board of Patent Appeal and Interferences has recently confirmed, a proper obviousness determination requires that an Examiner make “a searching comparison of the claimed invention - *including all its limitations* - with the teaching of the prior art.” See *In re Wada and Murphy*, Appeal 2007-3733, citing *In re Ochiai*, 71 F.3d 1565, 1572 (Fed.

Cir. 1995) (emphasis in original). “If an independent claim is nonobvious under 35 U.S.C. 103, then any claim depending therefrom is nonobvious” (MPEP §2143.03, citing *In re Fine*, 837 F.2d 1071, 5 USPQ2d 1596 (Fed. Cir. 1988)).

Accordingly, the Appellant submits that claim 1 is patentably distinct and non-obvious over AAPA and Lee for at least the reasons described above. Furthermore, independent claims 7, 12, 18, 22 and 25 are patentably distinct and non-obvious over the cited references, as claims 7, 12, 18, 22 and 25 cite features that are similar to the features of claim 1. In addition, claims 2-6, 8-11, 13-17, 19-21, 23-24 and 26-28 are patentably distinct and non-obvious over the references due at least to their dependencies from independent claims 1, 7, 12, 18, 22 and 25. As such, the Appellant respectfully requests the withdrawal of the rejection.

**2. CLAIMS 3, 5, 8, 10, 14, 16, 20-21, 23-24, AND 27-28 ARE
IMPROPERLY REJECTED BY THE USPTO UNDER 35 U.S.C.
§103(a) AS BEING UNPATENTABLE OVER AAPA, LEE AS
APPLIED TO CLAIM 1, IN VIEW OF BOSNAK.**

Claims 3, 5, 8, 10, 14, 16, 20-21, 23-24, and 27-28 have been rejected under 35 U.S.C. § 103 as being unpatentable over AAPA and Lee as applied to claim 1, in view of Bosnak.

As recited above, the Appellant submits that AAPA and Lee, alone or in any allowable combination absolutely fail to teach or suggest the Appellant’s claims 1-28 for at least the reasons recited above.

The Appellant submits however that the teachings of Bosnak absolutely fail to bridge the substantial gap between the teachings of AAPA and Lee and the invention of the Appellant. More specifically, the Appellant submits that Bosnak teaches “the operational status of a remotely controlled electronic siren is periodically tested, from a command post, without producing audible sound. The test procedure includes energizing the voice coils of the siren loudspeakers with a signal outside of the audible range, sensing whether current flows in the speaker voice coil circuits and storing the results of the test. The stored information, upon request, will be transmitted back to the command post.” (See Bosnak, Abstract).

The Appellant submits, however, that Bosnak absolutely fails to teach or suggest incrementally increasing/ decreasing the gain adjustments, a “means for **enabling the microphone**

output signal during first increments of time when the reproduced audio program signal is substantially off” and that “the microphone output signal includes ambient noise signal components without including reproduced audio program signal components” and “a signal process, in communication with the means for receiving and the means for enabling/disabling, for applying a first transfer function to the reproduced audio program signal, the first transfer function incrementally increasing gain adjustments to the reproduced audio program signal as a function of an increasing average amplitude of the microphone output signal over a period of time during which said microphone output signal is enabled, and incrementally decreasing gain adjustments to the reproduced audio program signal as a function of a decreasing average amplitude of the microphone output signal over a period of time during which said microphone output signal is enabled” as taught in the Appellant’s Specification and as claimed by the Appellant’s claims.

Accordingly, the Appellant submits that claim 1 is patentably distinct and non-obvious over AAPA, Lee and Bosnak for at least the reasons described above. Furthermore, independent claims 7, 12, 18, 22 and 25 are patentably distinct and non-obvious over the cited references, as claims 7, 12, 18, 22 and 25 cite features that are similar to the features of claim 1. In addition, claims 2-6, 8-11, 13-17, 19-21, 23-24 and 26-28 are patentably distinct and non-obvious over the references due at least to their dependencies from independent claims 1, 7, 12, 18, 22 and 25. As such, the Appellant respectfully requests the withdrawal of the rejection.

3. CLAIMS 4, 6, 9, 11, 15 AND 17 ARE IMPROPERLY REJECTED BY THE USPTO UNDER 35 U.S.C. §103(a) AS BEING UNPATENTABLE OVER AAPA, LEE AND BOSNAK AS APPLIED TO CLAIM 3, IN VIEW OF TANAKA.

Claims 4, 6, 9, 11, 15 and 17 have been rejected under 35 U.S.C. § 103 as being unpatentable over AAPA, Lee and Bosnak as applied to claim 3, in view of Tanaka.

As recited above, the Appellant submits that AAPA, Lee and Bosnak, alone or in any allowable combination absolutely fail to teach or suggest the Appellant’s claims 1-28 for at least the reasons recited above.

The Appellant submits however that the teachings of Tanaka absolutely fail to bridge the substantial gap between the teachings of AAPA, Lee and Bosnak and the invention of the Appellant. More specifically, the Appellant submits that Tanaka teaches a bass reproduction speaker apparatus,

which “includes: a cabinet with an opening, having a division member inside thereof; a speaker unit disposed at the division member; a passive radiator disposed in the opening; an amplifier for driving the speaker unit; a detector for detecting a vibration of a moving system of the speaker unit; and a feedback circuit for feeding back an output signal from the detector to the amplifier.” (Tanaka Abstract).

The Appellant submits, however, that Tanaka absolutely fails to teach or suggest incrementally increasing/ decreasing the gain adjustments, a “means for **enabling the microphone output signal** during first increments of time **when the reproduced audio program signal is substantially off**” and that “the microphone output signal includes ambient noise signal components without including reproduced audio program signal components” and “a signal process, in communication with the means for receiving and the means for enabling/disabling, for applying a first transfer function to the reproduced audio program signal, the first transfer function incrementally increasing gain adjustments to the reproduced audio program signal as a function of an increasing average amplitude of the microphone output signal over a period of time during which said microphone output signal is enabled, and incrementally decreasing gain adjustments to the reproduced audio program signal as a function of a decreasing average amplitude of the microphone output signal over a period of time during which said microphone output signal is enabled” as taught in the Appellant’s Specification and as claimed by the Appellant’s claims.

Accordingly, the Appellant submits that claim 1 is patentably distinct and non-obvious over AAPA, Lee, Bosnak and Tanaka for at least the reasons described above. Furthermore, independent claims 7, 12, 18, 22 and 25 are patentably distinct and non-obvious over the cited references, as claims 7, 12, 18, 22 and 25 cite features that are similar to the features of claim 1. In addition, claims 2-6, 8-11, 13-17, 19-21, 23-24 and 26-28 are patentably distinct and non-obvious over the references due at least to their dependencies from independent claims 1, 7, 12, 18, 22 and 25. As such, the Appellant respectfully requests the withdrawal of the rejection.

Conclusion

In light of these remarks, it is submitted that claims 1-28 would not have been obvious to a person of ordinary skill in the art upon a reading of AAPA in view of Lee, Bosnak and Tanaka, whether taken separately or in combination. Therefore, it is believed that claims 1-28 are allowable under 35 U.S.C. §103. It is respectfully requested that the Board of Patent Appeals and Interferences reverse the rejection of claims 1-28.

Respectfully submitted,

Date: **March 28, 2012**

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VIII. CLAIMS APPENDIX

1. (Previously Presented) A speaker system providing enhanced intelligibility of a reproduced audio program signal in the presence of ambient noise, the speaker system comprising:

means for receiving the reproduced audio program signal;

a microphone for monitoring at least ambient noise signals and for providing a microphone output signal;

means for enabling the microphone output signal during first increments of time when the reproduced audio program signal is substantially off, and disabling the microphone output signal during second increments of time when the reproduced audio program signal is on, such that the microphone output signal includes ambient noise signal components without including reproduced audio program signal components; and

a signal process, in communication with the means for receiving and the means for enabling/disabling, for applying a first transfer function to the reproduced audio program signal, the first transfer function incrementally increasing gain adjustments to the reproduced audio program signal as a function of an increasing average amplitude of the microphone output signal over a period of time during which said microphone output signal is enabled, and incrementally decreasing gain adjustments to the reproduced audio program signal as a function of a decreasing average amplitude of the microphone output signal over a period of time during which said microphone output signal is enabled.

2. (Original) The speaker system according to claim 1, wherein the incremental gain adjustments are in steps of between 1 dB and about 10 dB.

3. (Previously Presented) The speaker system according to claim 1, further comprising a first amplifier having an input and an output, the first amplifier input coupled to the output signal of the signal processor and the first amplifier output coupled to an input of a first speaker.

4. (Original) The speaker system according to claim 3, wherein the first speaker comprises a single speaker driver having a diaphragm diameter not greater than about 100 centimeters (cm).

5. (Previously Presented) The speaker system according to claim 3, further comprising:

a low-pass filter having an input and an output, the filter input coupled to the output signal of the signal processor and the filter output augmenting the first speaker output in a low frequency region; and

a second amplifier having an input and output, the second amplifier input coupled to the filter output and the second amplifier output coupled to an input of a second speaker.

6. (Original) The speaker system according to claim 5, wherein the first and second speakers each comprise a single speaker driver having a diaphragm diameter not greater than about 100 centimeters (cm).

7. (Previously Presented) A speaker system providing enhanced intelligibility of a reproduced audio program signal in the presence of ambient noise, the speaker system comprising:

means for receiving the reproduced audio program signal;

a microphone for monitoring ambient noise signals and for providing a microphone output signal;

means for enabling the microphone output signal during first increments of time when the reproduced audio program signal is substantially off, and disabling the microphone output signal during second increments of time when the reproduced audio program signal is on, such that the microphone output signal includes ambient noise signal components without including reproduced program signal components; and

a signal processor, in communication with the means for receiving and the means for enabling/disabling for applying a transfer function to the reproduced audio program signal, the transfer function incrementally increasing high frequency response of the reproduced audio program signal as a function of a decreasing amplitude of the microphone output signal averaged over a period of time during which said microphone output signal is enabled, and vice versa, wherein the signal process output signal is maintained during such times as the microphone output signal is disabled.

8. (Previously Presented) The speaker system according to claim 7, further comprising a first amplifier having an input and an output, the first amplifier input coupled to the output signal of the signal processor and the first amplifier output coupled to an input of a first speaker.

9. (Original) The speaker system according to claim 8, wherein the first speaker comprises a single speaker driver having a diaphragm diameter not greater than about 100 centimeters (cm).

10. (Previously Presented) The speaker system according to claim 8, further comprising:

a low-pass filter having an input and an output, the filter input coupled to the output signal of the signal processor and the filter output augmenting the first speaker output in a low frequency region; and

a second amplifier having an input and output, the second amplifier input coupled to the filter output and the second amplifier output coupled to a second speaker.

11. (Original) The speaker system according to claim 10, wherein the first and second speakers each comprise a single speaker driver having a diaphragm diameter not greater than about 100 centimeters (cm).

12. (Previously Presented) A speaker system providing enhanced intelligibility of a reproduced audio program signal in the presence of ambient noise, the speaker system comprising:

means for receiving the reproduced audio program signal;

a microphone for monitoring ambient noise signals and for providing a microphone output signal;

means for enabling the microphone output signal during first increments of time when the reproduced audio program signal is substantially off, and disabling the microphone output signal during second increments of time when the reproduced audio program signal is on, such that the microphone output signal includes ambient noise signal components without including reproduced program signal components; and

a signal processor, in communication with the means for receiving and the means for enabling/disabling, including a first transfer function and a second transfer function having at

least one signal processor output signal, wherein:

the first transfer function provides incrementally increasing gain adjustments to the reproduced audio program signal as a function of an increasing average amplitude of the microphone output signal over a period of time during which said microphone output signal is enabled, and vice versa;

the second transfer function provides incrementally increasing high frequency response of the reproduced audio program signal as a function of a decreasing average amplitude of the microphone output signal over a period of time during which said microphone output signal is enabled, and vice versa; and

the at least one signal processor output signal is maintained during such time as the microphone output signal is disabled.

13. (Original) The speaker system according to claim 12, wherein the incremental gain adjustments are in steps of between about 1 dB and about 10 dB.

14. (Previously Presented) The speaker system according to claim 12, further comprising a first amplifier having an input and an output, the first amplifier input coupled to the at least one output signal of the signal processor and the first amplifier output coupled to a first speaker.

15. (Original) The speaker system according to claim 14, wherein the first speaker comprises a single speaker driver having a diaphragm diameter not greater than about 100 centimeters (cm).

16. (Previously Presented) The speaker system according to claim 14, further comprising:

a low-pass filter having an input and an output, the filter input coupled to the at least one output signal of the signal processor and the filter output augmenting the first speaker output in a low frequency region; and

a second amplifier having an input and output, the second amplifier input coupled to the filter output and the second amplifier output coupled to a second speaker input of a second speaker.

17. (Original) The speaker system according to claim 16, wherein the first and second speakers each comprise a single speaker driver having a diaphragm diameter not greater than about 100 centimeters (cm).

18. (Previously Presented) A method of enhanced intelligibility of a reproduced audio program signal in the presence of ambient noise in a speaker system comprising:

receiving the reproduced audio program signal;

monitoring ambient noise signals using a microphone to provide a microphone output signal;

enabling the microphone output signal during first increments of time when the reproduced audio program signal is substantially off, and disabling the microphone output signal during second increments of the time when the reproduced audio program signal is on, such that the microphone output signal includes ambient noise signal components without including reproduced program signal components; and

processing the reproduced audio program signal and the microphone output signal using a first transfer function, the first transfer function having a signal process output signal, the first transfer function providing incrementally increasing gain adjustments to the reproduced audio program signal as a function of an increasing average amplitude of the microphone output signal over a period of time during which said microphone output signal is enabled, and incrementally decreasing gain adjustments to the reproduced audio program signal as a function of a decreasing average amplitude of the microphone output signal over a period of time during which said microphone output signal is enabled.

19. (Original) The method according to claim 18, wherein the incremental gain adjustments are in steps of between about 1 dB and about 10 dB.

20. (Previously Presented) The method according to claim 18, further comprising:

amplifying the signal process output signal using a first amplifier to produce a first amplified output signal; and

coupling the first amplified output signal to a first speaker input of a first speaker.

21. (Previously Presented) The method according to claim 20, further comprising:

filtering the signal process output signal using a low-pass filter to produce a filtered output signal;

amplifying the filtered output signal using a second amplifier to reproduce a second amplified output signal; and

coupling the second amplified output signal to an input of a second speaker.

22. (Previously Presented) A method of enhanced intelligibility of a reproduced audio program signal in the presence of ambient noise in a speaker system, the method comprising:

receiving the reproduced audio program signal;

monitoring ambient noise signals using a microphone to provide a microphone output signal;

enabling the microphone output signal during first increments of time when the reproduced audio program signal is substantially off, and disabling the microphone output signal during second increments of time when the reproduced audio program signal is on, such that the microphone output signal includes ambient noise signal without including reproduced program signal components; and

processing the reproduced audio program signal and the microphone output signal using a second transfer function, the second transfer function providing incrementally increasing high frequency response of the reproduced audio program signal as a function of a decreasing average amplitude of the microphone output signal over a period of time during which said microphone output signal is enabled, and vice versa, wherein the signal process output signal is maintained during such times as the microphone output signal is disabled.

23. (Previously Presented) The method according to claim 22, further comprising:

amplifying the signal process output signal using a first amplifier to produce a first amplified output signal; and

coupling the first amplified output signal to a first speaker input of a first speaker.

24. (Previously Presented) The method according to claim 23, further comprising:

- filtering the signal process output signal using a low-pass filter to produce a filtered output signal;

- amplifying the filtered output signal using a second amplifier to produce a second amplified output signal; and

- coupling the second amplified output signal to an input of a second speaker.

25. (Previously Presented) A method of enhanced intelligibility of a reproduced audio program signal in the presence of ambient noise in a speaker system comprising:

- receiving the reproduced audio program signal;

- monitoring ambient noise signals using a microphone to provide a microphone output signal;

- enabling the microphone output signal during first increments of time when the reproduced audio program signal is substantially off, and disabling the microphone output signal during second increments of time when the reproduced audio program signal is on, such that the microphone output signal includes ambient noise signal components without including reproduced program signal components; and

- processing the reproduced audio program signal and the microphone output signal using a first transfer function and a second transfer function, the first and second transfer functions having at least one signal process output signal, wherein:

- the first transfer function provides incrementally increasing gain adjustments to the reproduced audio program signal as a function of an increasing average amplitude of the microphone output signal over a period of time during which said microphone output signal is enabled, and vice versa;

- the second transfer function provides incrementally increasing high frequency response of the reproduced audio program signal, and vice versa; and

- the least one signal process output signal is maintained during such times as the microphone output signal is disabled.

26. (Original) The method according to claim 25, wherein the incremental gain adjustments are in steps of between about 1 dB and about 10 dB.

27. (Previously Presented) The method according to claim 25, further comprising:
 amplifying the at least one signal; and
 coupling the first amplified output signal to a first speaker input of a first speaker.

28. (Previously Presented) The method according to claim 27, further comprising:
 filtering the at least one signal process output signal using a low-pass filter to produce a filtered output signal;
 amplifying the filtered output signal using a second amplifier to produce a second amplified output signal; and
 coupling the second amplified output signal to a second speaker input of a second speaker.

IX. EVIDENCE APPENDIX

No evidence has been submitted pursuant to §§ 1.130, 1.131, or 1.132 of this title. No other evidence has been entered by the Examiner and/or relied upon by Appellant in this appeal, at this time.

X. RELATED PROCEEDINGS APPENDIX

Appellant is not aware of any appeals or interferences related to the present application.